COMMUNICATION METHOD AND APPARATUS FOR MULTI-USER DETECTION

Field of the Invention

The present invention relates to a method, apparatus and computer program for the iterative acquisition of signals for multi-user detection and decoding.

Background

Multi-user detection and decoding (MUD) techniques allow detection and decoding of transmissions by two or more mutually interfering users over an interference channel.

In general, MUD involves a detection process in which the received composite signal is resolved into symbol estimates for each user, and a decoding process in which the symbols are decoded to recover their data content, using a forward error correction (FEC) decoding algorithm. An optimal joint decoder combines these two processes using a maximum likelihood decoding technique to minimize the probability of decoder error. This technique has a complexity that increases exponentially with the number of users and the FEC codeword length, and may not be possible with certain FEC coding techniques such as Turbo codes. Hence, it is necessary to devise sub-optimal techniques with reduced complexity.

One sub-optimal approach is to separate the detection and decoding processes. Joint decisions are made on each symbol in the detection process, and the symbol streams are then independently decoded using conventional decoding techniques.

Another approach is iterative decoding, where soft decisions by the detector are input to separate decoders for each user, and the soft decisions by the decoders are fed back iteratively to the detector. Examples of iterative decoding algorithms are disclosed in 'An Iterative Multiuser Decoder for Near-Capacity Communications', Moher M, IEEE Transactions on Communications vol. 46, No. 7, July 1998 and 'Multiuser Decoding for Multibeam systems', Moher M, IEEE Transactions on Vehicular Technology, July 2000, Volume 49, Number 4, pages 1226-1234.

Before detection and decoding can take place, the timing, frequency and phase offset of each transmission must be acquired. Timing acquisition is particularly important, because frequency and phase estimation depend on the correct timing being acquired. Moreover, if a signal cannot be acquired or is incorrectly acquired, it cannot be decoded.

Statement of the Invention

According to one aspect of the present invention, there is provided a method of iteratively acquiring the timings of a plurality of transmissions in a signal received over a multiple access interference channel, the method comprising: estimating the relative timings of each of the transmissions; separately soft demodulating, decoding and remodulating each of the transmissions to generate soft estimates of each of the transmissions; for each transmission, cancelling the soft estimates of the other ones of the transmissions to generate an updated estimate of that transmission, and estimating the relative timings of each of the updated estimates of the transmissions.

Other aspects of the present invention include a computer program for performing the method, and apparatus arranged to perform the method.

Brief Description of the Drawings

Specific embodiments of the present invention will now be described with reference to the accompanying drawings, in which:

Figure 1 shows the format of a burst to be acquired in an embodiment of the invention;

Figure 2 is a diagram of multiple transmitters transmitting bursts over a multiple access channel;

Figure 3 is a schematic diagram of a multi-user detector and decoder with an acquisition function in an embodiment of the present invention;

Figure 4 is a diagram of a differential detector for use in the acquisition function;

Figure 5 is a diagram of a coherent detector for use in the acquisition function;

Figure 6 is a graph illustrating acquisition performance in a first simulation of the embodiment;

Figure 7 is a graph illustrating acquisition performance in a second simulation of the embodiment; and

Figure 8 is a chart showing regions in which acquisition can be achieved in the embodiment.

Detailed Description of the Embodiments

Burst Format

Figure 1 shows one example of a format of transmitted bursts to be acquired in an embodiment of the present invention. The burst B comprises an initial unique word UW1, data D, and a final unique word UW2. The unique words are predetermined bit sequences, having low auto-correlation, which are known by a receiver and can therefore be used for burst acquisition. The presence of the final unique word UW2 is not essential, but use of both unique words improves acquisition performance.

The data D comprises a sequence of modulated symbols x[i], as will be described in more detail below.

A preamble or control word (not shown) may be transmitted before the initial unique word UW1, and a guard interval may be left between consecutive bursts in the same frequency channel.

As one specific example, the bursts B may be MESP5 or MESP20 packets complying with the Inmarsat™ MPDS (mobile packet data service) specification, as follows:

Table 1 – MESP5 Packet Format

Modulation	16-QAM
Input bits per burst	192
Coding rate	3/7
Output bits per burst	448
Output symbols per burst	112
Preamble	4
Initial UW (symbols)	20
Final UW (symbols)	20
Total symbols /5 ms slot	156
Guard Time (symbols)	12
Symbol rate (ksps)	33.6
Slot length	5 ms

Table 2 – MESP20 Packet Format

Modulation	16-QAM
Input bits per burst	1192
Coding rate	½
Output bits per burst	2384
Output symbols per burst	596
Preamble	4
Initial UW (symbols)	40
Final UW (symbols)	20
Total symbols /5 ms slot	660
Guard Time (symbols)	12
Symbol rate (ksps)	33.6
Slot length	20 ms

Transmitter

Figure 2 shows a plurality K of users outputting respective bit sequences $b_1[i] \ldots b_K[i]$ encoded by encoders $C_1 \ldots C_K$ to produce coded sequences $d_1[i] \ldots d_K[i]$, which are interleaved by respective interleavers $\Pi_1 \ldots \Pi_K$ to generate interleaved sequences $\Pi_1(d_1[i]) \ldots \Pi_K(d_K[i])$, which are in turn modulated by modulators $M_1 \ldots M_K$ to generate the respective sequences of modulated symbols $x_1[i] \ldots x_K[i]$ at time i. The modulated symbols are transmitted in bursts such as shown in Figure 1. The data portion D preferably contains an integral number of blocks encoded by the encoders $C_1 \ldots C_K$, and the encoders are reset after each block, so that the encoding of one burst is independent of the content of any other burst.

In one specific example, the encoders $C_1 \ldots C_K$ are Turbo encoders i.e. parallel systematic recursive convolutional encoders, one or more but not all of which have an interleaver at the input, as described for example in 'Near Shannon limit error-correcting

coding and decoding: Turbo codes', Berrou, C., Glavieux, A. and Thitimajshima, P, *Proc. of ICC '93*, pp 1064-1070. The modulators $M_1 \ldots M_K$ may be 16QAM modulators, as used for example in the Inmarsat™ MPDS.

Multiple Access Channel

The modulated symbol sequences $x_1[i] \ldots x_K[i]$ are transmitted on a multiple access channel MA such that a set of symbol sequences $y_1[i] \ldots y_K[i]$ are received at a receiver. The effect of the multiple access channel MA may be modelled as:

$$\mathbf{y}[i] = \mathbf{AWx}[i] + \mathbf{n}[i] \qquad (1)$$

where $\mathbf{y}[i]$ is the complex vector channel output,

$\mathbf{A}$ is a normalised correlation matrix representing the cross-correlation between symbol sequences,

$\mathbf{W}$ is a diagonal matrix representing the amplitudes of each user and

$\mathbf{n}[i]$ represents the channel noise.

Receiver Architecture

The received signal $y_1[i] \ldots y_K[i]$ is detected and decoded by an iterative MUD receiver as shown in Figure 3. A multi-user detector DET takes as its input the output of the multiple access channel MA and the current soft estimates (initialised to zero at the first iteration) of each user's average contribution to the received signal, subject to the current probability distributions on the data. The detector DET outputs updated soft estimates for each user by subtracting the current soft estimates of all the interfering users.

The soft estimates for the respective users are soft demodulated by soft demodulators $DEM_1 \ldots DEM_K$, which calculate the posterior probabilities of each possible symbol of the modulation constellation. For example, with a 16QAM scheme, for each input symbol a probability is calculated of that symbol being each of the possible 16 symbols of the constellation. The corresponding soft detected bits are reordered by deinterleavers (not shown, for clarity) and input to soft decoders $DEC_1 \ldots DEC_K$ which refine the probabilities of the coded bits derived from the soft demodulators $DEM_1 \ldots DEM_K$ by taking into account the knowledge of the FEC code. The bits are reordered once again by respective interleavers (not shown) and output to soft modulators $M_1 \ldots M_K$, which produce conditional expectations of the coded and modulated symbols according to the posterior probabilities calculated by the decoders $DEC_1 \ldots DEC_K$. These

average symbols are input to a model of an estimated multiple access channel EMA which updates the channel estimates for each user, on the basis of estimated channel parameters derived by an acquisition function ACQ, and feeds these back to the multi-user detector DET for the next iteration.

For each iteration of the MUD receiver algorithm, an acquisition function ACQ receives the estimates of each user's contribution to the channel from the multi-user detector DET and performs an acquisition algorithm, as will be described below, on each of the estimates. The time, frequency and phase detected for each of the users is output to the multi-user detector DET for use in the next iteration. At the first iteration, the multi-user detector DET has no knowledge of the users' contributions to the multiple access channel, so its outputs are simply equal to its inputs. The detected time, frequency and phase are also output to the estimated multiple access channel EMA.

Each user's contribution is identified by its acquired characteristics, such as timing and optionally frequency and phase, and each separate 'arm' of the MUD receiver operates on the updated soft estimate for a respective user with the acquired characteristics. As there are no current soft estimates during the first iteration of the MUD receiver, no cancellation is performed and each arm operates on the same received signal, but with the acquired characteristics of the respective user.

Provided that at least one user is acquired at the first iteration of the MUD receiver algorithm, the contribution of that user is subtracted from the contribution of the weaker users in subsequent iterations, thereby improving the likelihood of successful acquisition of the weaker users in subsequent iterations. Hence, the iterative acquisition technique is particularly suitable for acquiring weak users in the presence of interference from stronger users.

If a false acquisition of a user is made by the acquisition function ACQ, the following MUD iteration will attribute a low probability to the decoded signal from that user, which will lead to a very low weighting of that user's contribution by the multi-user detector DET. Hence, a false acquisition will have little effect on the acquisition and decoding of other users. In subsequent iterations, that user may be acquired correctly once the estimates of the other users have improved.

The MUD iterations are repeated, on the same received signal y1[i]...yK[i], a number of times determined by the desired decoding performance and the acceptable processing delay. For example, the number of MUD iterations per received signal set may

be fixed at a number likely to give the desired performance under most conditions. Alternatively, the MUD iterations may be repeated until the desired decoding accuracy is achieved for one or more of the users – this may be determined by the probabilities output by the soft decoders DEC1...DECK exceeding a predetermined threshold – subject to a maximum number of iterations or maximum processing delay. The decoded bits for each user are then output by the MUD receiver.

The receiver architecture may be implemented in software, programmed for example into a digital signal processor (DSP) or other hardware or firmware, which may form part of a terminal. The functional blocks shown in Figure 3 do not necessarily correspond to discrete hardware components.

Receiver Functional Description

Multi-user detector

The multi-user detector DET has the following inputs:

$\mathbf{y}[i]$	a sequence of complex K-vectors (for K users) received from the multiple access channel EMA
$\{\hat{\mathbf{y}}_1[i],\ldots\hat{\mathbf{y}}_K[i]\}$	a set of sequences of complex K-vectors from the estimated channel EMA
$\mathbf{M}[i]=\begin{bmatrix}\mathbf{m}[i]_1\\ \vdots\\ \mathbf{m}[i]_K\end{bmatrix}$	a sequence of complex $K \times K$ matrices. The row vector $\mathbf{m}[i]_k$ is a filter to be used for user k.

The multi-user detector DET has the following outputs:

$\hat{\mathbf{x}}[i]$	a sequence of complex K-vectors. The component $\hat{\mathbf{x}}_k[i]$ is the new estimate of symbol *i* for user *k*.
σ_k^2	estimated residual interference plus noise variance for user *k*.

The multi-user detector DET has the following operation:

For a synchronous channel,

$$\hat{\mathbf{x}}_k[i]=\left\langle \mathbf{m}_k[i],\left(\mathbf{y}[i]-\sum_{j\neq k}\hat{\mathbf{y}}_j[i]\right)\right\rangle \qquad (2)$$

This is the inner product between the filter for the user *k* and the results of cancelling all the estimated components other than user *k* from the channel output. For an

asynchronous channel, each sequence $\hat{x}_k[i]$ is delayed and equalised to compensate for symbol timing and frequency offset, as determined for example by the acquisition function ACQ.

The sequence of matrices M[i] represents a time-varying matrix filter. In the simplest case, this may be a diagonal matrix of complex values which representing gain and phase. These are assumed to be slowly varying and are estimated from unique words at the beginning and end of a burst.

The residual plus noise interference for each user is estimated from the completely cancelled signal:

$$\hat{n}_k[i] = \left\langle \mathbf{m}_k[i], \left(\mathbf{y}[i] - \sum_{j=1}^{K} \hat{\mathbf{y}}_j[i] \right) \right\rangle \qquad (3)$$

using a variance estimation technique:

$$\sigma_k^2 = \frac{1}{L-1} \sum_{i=1}^{L} \left(\hat{n}_k[i] - \frac{1}{L} \sum_{j=1}^{L} \hat{n}_k[j] \right)^2 \qquad (4)$$

where L is the number of symbols in the bursts.

Soft Demodulator

Each soft demodulator DEM_k has the following inputs:

$\hat{x}_k[i]$	sequence of complex numbers corresponding to the soft symbol estimates for user *k*
$Q = (Q_0, \ldots Q_{15})$	An ordered set containing the normalised complex constellation values (assuming QAM, otherwise the cardinality will be different)
$\{S^b : b = 0,1,2,3\}$	A set containing the indices of the constellation points in Q that correspond to bit b being equal to zero.
$(\sigma_1^2 \ldots \sigma_K^2)$	Noise plus residual interference estimates for each user

the following outputs:

$\left(p_k^0[i], p_k^1[i], p_k^2[i], p_k^3[i]\right)$	A sequence of 4-vectors (in the case of QAM). Element $p_k^b[i]$ is the prior probability, between 0 and 1, that bit b of symbol i for user k is zero

and calculates the bit probabilities as follows:

$$p_k^b[i] = \sum_{j \in S^b} \eta(\hat{x}_k[i] - Q_j) \tag{5}$$

where η is the zero mean, variance σ_k^2 Gaussian probability density function.

Channel Interleaver

Each channel interleaver and puncturer receives as input the data and parity bits from the corresponding coder C_k and interleaves and punctures them to generate sets of bits each corresponding to one symbol for modulation. This type of interleaver and puncturer is used in the Inmarsat™ IPDS. Each user uses the same interleaving and puncturing pattern.

Soft Decoder

Each soft decoder DEC_k has the following inputs:

N_C	The number of decoding iterations to perform for each MUD iteration
$\Pi_{k,\text{ CHANNEL}}$	The channel interleaver for the user k
$\Pi_{k,\text{ TURBO}}$	The Turbo interleaver for the user k
$\left(p_k^0[i], p_k^1[i], p_k^2[i], p_k^3[i]\right)$	A sequence of 4-vectors (in the case of QAM). Element $p_k^b[i]$ is the prior probability, between 0 and 1, that bit b of symbol i for user k is zero
$\pi_k[i]$	A sequence of scalars representing the prior probability that bit i for user k is zero

and the following outputs:

$\left(p_k^0[i], p_k^1[i], p_k^2[i], p_k^3[i]\right)'$	A sequence of 4-vectors (in the case of QAM). Element $p_k^b[i]$ is the posterior probability, between 0 and 1, that bit b of symbol i for user k is zero
$\pi_k[i]$	A sequence of scalars representing the posterior probability that bit i for user k is zero

The posterior coded and uncoded bit probabilities are calculated from the prior coded and uncoded bit probabilities using an iterative "soft-in/soft-out" Turbo decoder. Techniques for iterative Turbo decoding are well-known in the art, for example: 'Iterative Decoding of Binary Block and Convolutional Codes', Hagenauer J, IEEE Transactions on Information Theory, Vol. 42, No. 2, March 1996.

Soft Modulator

Each soft modulator M_k has the following inputs:

$Q = (Q_0, \ldots Q_{15})$	An ordered set containing the normalised complex constellation values (assuming QAM, otherwise the cardinality will be different)
$\{B_j : j = 0,1,\ldots 15\}$	A set containing the indices of the bits in j which are equal to zero. For example, B_0 is the set {0, 1, 2, 3} while B_5 is the set {1, 3}
$\left(p_k^0[i], p_k^1[i], p_k^2[i], p_k^3[i]\right)$	A sequence of 4-vectors (in the case of QAM). Element $p_k^b[i]$ is the posterior probability, between 0 and 1, that bit b of symbol i for user k is zero

and the following output:

$\hat{x}[i]$	a sequence of complex scalars representing the new soft estimate of symbol i for user j

The estimates are calculated as the expectation of the signal constellation according to the current symbol probabilities:

$$\hat{x}_k[i] = \sum_{j=0}^{15} Q_j P_j \qquad (6)$$

where the symbol probabilities are calculated according to

$$P_j = \left[\prod_{b \in B_j} p_k^b[i]\right]\left[\prod_{b \in \bar{B}_j} (1 - p_k^b[i])\right] \qquad (7)$$

where $\bar{B}_j$ is the complement set (i.e. the indices of the bits that are ones).

Acquisition Function

The acquisition function ACQ has the following input:

$\hat{x}[i]$	a sequence of complex scalars. $\hat{x}_k[i]$ represents the new soft estimate of symbol i for user k.

and outputs the current estimated channel parameters (phase, frequency, timing and amplitude). The acquisition function estimates the channel parameters for each of the sequences corresponding to the different users.

Estimated Multiple Access Channel

The estimated multiple access channel EMA has the following inputs:

$\hat{x}[i]$	a sequence of complex scalars. $\hat{x}_k[i]$ represents the new soft estimate of symbol i for user k
t_k, f_k, φ_k	The current estimated channel parameters, including time, frequency and phase estimates for each user.

and the following outputs:

$\{\hat{y}_1[i],\ldots\hat{y}_K[i]\}$	a set of sequences of complex K-vectors. The vector sequence $\hat{y}_k[i]$ corresponds to the current estimated contribution of user k to the channel output.

The estimated multiple access channel EMA models the effect of the actual multiple access channel according to the current estimates of the channel parameters. For example, if the channel is symbol synchronous,

$$\hat{y}_k[i] = \begin{pmatrix} \hat{w}_{1k} \\ \hat{w}_{2k} \\ \vdots \\ \hat{w}_{Kk} \end{pmatrix} \hat{x}_k[i] \qquad (8)$$

where $\hat{w}_{jk}$ is the estimated complex gain from user k to output j.

Acquisition Algorithms

Differential Detection

In one specific embodiment, a differential detection algorithm is used in the acquisition function ACQ. As represented in Figure 4, the algorithm takes the initial and final unique words UW1 and UW2 of the relevant burst B as input and performs a time offset estimation ACQ_t. The estimated offset τ is then provided as input to a frequency

estimation stage ACQ_f which calculates a frequency offset f using only the initial unique word UW1. The frequency estimation stage ACQ_f will not be described further.

In this algorithm, the time offset is detected by the use of differential correlation between the received burst and the reference value or values of the initial and final unique words UW1, UW2; this method is possible because of the low auto-correlation of the unique words.

The unique words UW1 and UW2 may be constant for all bursts, or may be selected from one of a plurality of possible unique words – this technique has various uses which will not be described here. In the latter case, correlation may be performed between the received burst and each of the possible reference unique words, and the reference unique word having the highest correlation peak is determined to correspond to the unique word present in the received burst.

First, the received signal *y(t)* is scalar multiplied by the conjugate of the reference unique word $\hat{y}(t)$ with a test offset τ which is varied between zero and the maximum time offset in steps of one sample period:

$$s(t) = y(t).\hat{y}(t - \tau) \qquad (9)$$

The power R_0 of the result is then calculated as follows:

$$R_0 = s(t) \times s'(t) \qquad (10)$$

where s'(t) is the transpose of s(t). Next, the differential power R_1 is calculated:

$$R_1 = s(t - 4T_{symbol} + T_{sample}) \times s'(t + T_{sample}) \qquad (11)$$

where T_{sample} is the sample interval of the receiver and T_{symbol} is the symbol period. For example, in a four times oversampling receiver, $T_{symbol} = 4 \times T_{sample}$.

The ratio of R_1 to R_0 is calculated for each value of τ and the value of τ which gives the highest ratio is taken as the best estimate of the time offset of the received burst.

The following pseudocode describes the differential detection algorithm:

```
Start estimation
      Remove guard and control word from received packet
      FOR I=0;one sample steps; maximum offset
            Scalar multiply received signal and conjugate of reference
            UW
            Calculate the power of multiplication results
            Calculate differential power of multiplication results
            Calculate and record the ratio of the two powers and
            corresponding time offset
```

```
        Offset the reference UW by one sample
    END FOR
    Find maximum value of recorded power ratios
    Return the time offset corresponding to maximum value
End estimation
```

Differential detection has low complexity when detecting packets with large frequency offsets (e.g. > 500 Hz). However, a performance gain may be expected by using a computationally more complex coherent detection algorithm, for example as described below.

Coherent Detection

In an alternative embodiment, a coherent detection algorithm is used in the acquisition function ACQ. As represented in Figure 5, the algorithm uses the initial and final unique words UW1 and UW2 of the relevant burst B to perform coherent estimation of both time and frequency offset $ACQ_{t,f}$. The estimated time offset τ and frequency offset are then used by a phase estimation stage ACQ_{φ} which calculates a phase offset φ using only the initial unique word UW1. The phase estimation stage ACQ_{φ} will not be described further.

First, the frequency offset is estimated by extracting two windows of data, with length corresponding to that of the initial unique word UW1, from respectively the beginning and end of the burst. The first window is correlated with the reference initial unique word UW1 and the second window with the reference final unique word UW2. The correlation of each window with the reference UW is modelled by:

$$\begin{aligned} r(t) &= \hat{y}(t-\tau) * y(t) \\ &= \int_{-\infty}^{\infty} \hat{y}(\lambda-\tau)\, y(\lambda-\tau)\, d\lambda \qquad (12) \end{aligned}$$

where $y(t)$ is the received signal, $\hat{y}(t)$ is the reference UW and τ is the time offset applied in the current iteration, and which varies from zero to maximum offset in steps of one sample period of the receiver. The symbol '$*$' denotes correlation.

The fast Fourier transforms (FFT) of the correlations are then taken and their magnitudes summed to yield a vector which is peak picked to find its maximum value. The FFT is calculated according to:

$$R(k)=\frac{1}{N}\sum_{n=0}^{N-1}x(n)e^{-j\frac{2\pi nk}{N}} \qquad (13)$$

where N is the number of samples, n is the discrete representation of time t, and k is the discrete representation of frequency ω.

The extraction is done in steps of one sample from zero to the maximum time offset. The frequency where the highest peak lies is taken as the estimated frequency offset. The sample index of the highest peak corresponds to the estimated time offset.

The following pseudocode describes the implementation of the coherent estimation algorithm:

```
Start of estimation
     Remove guard and control word from received packet
     FOR I=0; one sample steps; maximum offset
          Correlate received signal with reference unique words
          FFT the correlation result
          Find and record highest FFT peak and its corresponding
          frequency
          Offset the reference unique words by one sample
     END FOR
     Find the maximum value among the recorded highest peaks
     Return the time offset and frequency corresponding to the maximum
     value
End estimation
```

Optimal Multi-user acquisition

Under high interference conditions, the single-user acquisition approaches described above (differential or coherent detection) may limit performance, and it may be preferable to use a multi-user acquisition algorithm. Such multi-user algorithms are generally too complex to evaluate in detail, but an outline of one possible approach is given below.

The following base band linear Gaussian signal model is assumed. K users transmit the same, known unique word x(t) to the receiver. The relative time offsets for the users are expressed as a vector $\boldsymbol{\tau} = (\tau_1, \tau_2 \ldots \tau_K)$. For simplicity, the data portion D is ignored and it is assumed that the signal x(t) is zero outside the interval of the unique word. The vector of received signals:

$$y_i(t)=\sum_{k=1}^{K}A_{ik}x(t-\tau_k)+n_i(t) \qquad (14)$$

where A_{ik} is the gain from the k^{th} user's transmitter to spot beam i in a multiple spot beam satellite system; $n_i(t)$ are circularly symmetric complex Gaussian noise with covariance $E[n_i(t)n_j^*(t+\tau)] = N_0$ if both $\tau = 0$ and $i = j$, and zero otherwise. For simplicity, frequency and phase offset are assumed to be zero.

If the channel gains A_{ik} are unknown at the receiver, then an optimal maximum likelihood detector makes a joint decision on the gains and the delays $\tau_1, \tau_2 \ldots \tau_K$, according to:

$$\hat{\mathbf{A}}, \hat{\boldsymbol{\tau}} = \arg\max_{\mathbf{A}, \boldsymbol{\tau}} p(\mathbf{y} \mid \mathbf{A}, \boldsymbol{\tau}) \tag{15}$$

Under the Gaussian assumption, choose the channel gains $\hat{\mathbf{A}}$ and delays $\hat{\boldsymbol{\tau}}$ that minimize the Euclidean distance between the received signal $\mathbf{y}$ and the hypothesized signal $\hat{\mathbf{y}}$, where:

$$\hat{y}_i(t) = \sum_{k=1}^{K} \hat{A}_{ik} x(t - \hat{\tau}_k) \tag{16}$$

Thus the optimal detector operates according to:

$$\hat{\mathbf{A}}, \hat{\boldsymbol{\tau}} = \arg\max_{\hat{\mathbf{A}}, \hat{\boldsymbol{\tau}}} \sum_{i=1}^{r} \int \left| y_i(t) - \hat{y}_i(t) \right|^2 dt \tag{17}$$

If the channel gains are known, then only the delays are estimated, i.e. the maximization is only performed over $\boldsymbol{\tau}$. The maximal likelihood detector essentially performs maximal ratio combining on the received signals, involving searching all possible combinations of user delays; this is prohibitively complex with currently available hardware, although may become feasible with advances in processor power. If phase and frequency offsets are also present, then all possible combinations of delay, frequency and phase offsets for all users would have to be searched.

Hence, whilst it may be possible to perform multi-user acquisition, single-user techniques are preferred.

Acquisition Simulation Results

The effects of iterative acquisition using coherent detection were simulated in a two-user scenario. A 'strong' user has a carrier to interference ration C/I = 10 dB and a 'weak' user is simulated using a range of smaller values of C/I. The thermal noise power E_S/N_0 is set the same for both users. Random user data D was created and encapsulated into bursts, with random timing offsets between 0 and 0.5 ms, but no frequency or phase

offsets, since the aim was to test timing acquisition performance only. The bursts were combined in a simulated multiple user channel and additive white Gaussian noise (AWGN) was added.

Tables 3 to 5 below show the results for simulation of 10,000 20ms frames, after a first and a second iteration of the MUD algorithm, with the weak user C/I = 1, 2 and 4 dB respectively.

Table 3 – Weak User C/I = 1dB

E_s/N_0	Strong User Errors	Weak User Iteration 1 Errors	Weak User Iteration 2 Errors
0	1	3100	31
1	0	2110	0
3	0	1500	0
6	0	990	0

Table 4 – Weak User C/I = 2dB

E_s/N_0	Strong User Errors	Weak User Iteration 1 Errors	Weak User Iteration 2 Errors
0	1	990	8
1	0	770	0
3	0	310	0
6	0	110	0

Table 5 – Weak User C/I = 4dB

E_s/N_0	Strong User Errors	Weak User Iteration 1 Errors	Weak User Iteration 2 Errors
0	1	790	5
1	0	220	0
3	0	46	0
6	0	6	0

Figure 6 shows the acquisition performance for the weak user before and after (shaded region) the strong user was subtracted. These results show that the iterative acquisition method can successfully acquire users that cannot be acquired by single-user

non-iterative techniques. For $E_S/N_0 \geq 1$ dB, the weak user can be acquired (error rate less than 10^{-4}) using the iterative method.

A further simulation was performed using Inmarsat™ IPDS bearer sub-type L3, which has a coding rate of 1/3, under fading conditions (C/M = 15 dB, fading bandwidth 20 Hz, multipath delay = 0). Otherwise, the parameters were the same as in the simulation above. The results are shown in Tables 6 to 8 below.

Table 6 – Weak User C/I = 1dB

E_s/N_0	Strong User Errors	Weak User Iteration 1 Errors	Weak User Iteration 2 Errors
1.25	1	4360	4
2.25	0	960	0
3.25	0	600	0

Table 7 – Weak User C/I = 2dB

E_s/N_0	Strong User Errors	Weak User Iteration 1 Errors	Weak User Iteration 2 Errors
1.25	0	2810	2
2.25	0	1300	0
3.25	0	525	0

Table 8 – Weak User C/I = 4dB

E_s/N_0	Strong User Errors	Weak User Iteration 1 Errors	Weak User Iteration 2 Errors
1.25	1	1100	2
2.25	0	620	0
3.25	0	320	0

The results are shown in Figure 7. Acquisition is again possible in regions where it would not be possible using single-user non-iterative techniques only. For $E_S/N_0 > 2.25$, the weak user can be successfully acquired.

Figure 8 shows the region in which the iterative acquisition method can be used successfully to acquire a weak user in the case where the strong user has C/I = 10 dB. The

horizontal axis represents the C/I for the weak user, while the vertical axis represents signal to noise ratio (E_S/N_0), which is common to both users. The shaded region above and to the right of the line represents the conditions where acquisition of the weak user is achievable with error rate less than 10^{-4}. Hence, the iterative method greatly improves the acquisition performance for weaker users.

In summary, it has been demonstrated that the use of iterative acquisition techniques in combination with iterative decoding techniques gives superior performance to non-iterative initial acquisition techniques in a multi-user detector. The superior performance is due at least in part to the use of information derived by a decoding iteration to improve the performance of the next acquisition iteration. This advantage is not dependent on the specific details of the examples and simulations described above. Hence, the skilled person will recognise that the present invention is not limited save as defined in the following claims.